The Claims

1) A method of audio (as defined herein) transmission over a network where audio frames are sent in UDP packets, wherein the audio frames are overlapped by at least one for each UDP packet.

2) A method as claimed in claim 1, wherein there are two audio frames and one overlapped audio frame for each UDP packet.

3) A method as claimed in claim 1, wherein there are two audio frames and two overlapped audio frames for each UDP packet.

4) A method as claimed in of claim 1, wherein the audio frames are overlapped in response to a detection of high packet loss.

5) A method as claimed in claim 4, wherein the extent of overlap is selected based on the extent of the packet loss.

6) A method as claimed in claim 5, wherein the overlapped audio frames are converted to non-overlapped audio format by an audio converter prior to being received at a terminating gateway, the audio converter being located close to the terminating gateway.

7) A method as claimed in claim 1, wherein the overlapped audio frames are converted to non-overlapped audio format by a terminating audio converter prior to being received at a terminating gateway, the terminating audio converter being located close to the terminating gateway.

8) A method as claimed in claim 1, wherein the transmission from an originating gateway is in a non-overlapped audio format and is to an originating audio converter to convert the transmission to overlapped format; the originating audio converter being close to the originating gateway.

9) A method as claimed in claim 6, wherein the transmission from an originating gateway is in a non-overlapped audio format and is to an originating audio converter to convert the transmission to overlapped format; the originating audio converter being close to the originating gateway.

10) A method as claimed in claim 7, wherein the transmission from an originating gateway is in a non-overlapped audio format and is to an originating audio converter to convert the transmission to overlapped format; the originating audio converter being close to the originating gateway.

11) A method as claimed in claim 8. wherein the originating audio converter is in the same network as the originating gateway.

12) A method as claimed in claim 7, wherein the terminating audio converter is in the same network as the terminating gateway.

13) A method of internet telephony on a network where audio (as defined herein) frames are sent in UDP packets, wherein at least one monitoring station is provided in the network, and wherein the at least one monitoring station periodically sends at least one packet to at least one destination of interest to obtain quality of service information, the quality of service information being used to dynamically alter call set-up.

14) A method as claimed in claim 13, wherein the at least one monitoring station performs a trace route analysis at required intervals.

15) A method as claimed in claim 13, wherein the quality of service information is periodically uploaded to at least one central side for amalgamation of data.

16) A method as claimed in claim 14, wherein the quality of service

information is periodically uploaded to at least one central site for amalgamation of data.

17) A method as claimed in claim 15, wherein the quality of service information obtained from a first packet sent to a first destination is compared with packet historical values for the first destination and, if a discrepancy above a predetermined threshold is obtained, a trace route analysis is performed for the first destination.

18) A method as claimed in claim 17, wherein the results of the trace route analysis are compared to trace route historical values and, if the results exceed a present level of discrepancy, traffic to the first destination can be sent over the network by a different route.

19) A method as claimed in claim 14, wherein the results of the trace route analysis are compared to trace route historical values and, if the results exceed a present level of discrepancy, traffic to the first destination can be sent over the network by a different route.

20) A method as claimed in claim 13, wherein the at least one packet has a set-up substantially the same as the set-up of a standard packet for voice over Internet protocol calls.

21) A method as claimed in claim 20, wherein the at least one packet is of the same size as those used for voice over Internet protocol calls.

22) A method as claimed in claim 20, wherein the at least one packet is in the same user data protocol port range as those used for voice over Internet protocol calls.

23) A method as claimed in claim 13, wherein there are a plurality of packets sent at a controlled rate to emulate the rate of packets of voice over Internet protocol packets.

24) A method as claimed in 17, wherein the first destination is capable of measuring jitter between packets arriving from the at least one monitoring station.

25) A method as claimed in claim 24, wherein the first destination keeps statistics on the packets it has received and forwards those statistics to the at least one monitoring station.

26) A method as claimed in claim 13, wherein the quality of service information obtained includes one or more selected from the group consisting of:

a) the maximum round trip time;

b) the number of packets lost;

c) the round trip jitter;

d) the number of packets received out of sequence; and

e) the number of consecutive packets lost.

27) A method as claimed in of claim 13, wherein prior to sending the at least one packet, the at least one monitoring station sends a call set-up request.

28) A packet to be used to provide quality of service information about a telecommunications network, the packet having a set-up substantially the same as the set-up of a standard packet for voice over Internet protocol calls.

29) A packet as claimed in claim 26, wherein the packet is of the same size as the standard packet.

30) A packet as claimed in claim 26, wherein the packet is in the same user data protocol port range as the standard packet.

31) A packet as claimed in claim 26, wherein the packet is sent over the telecommunications network to emulate the standard packet.

32) A packet as claimed in claim 26, wherein the packet emulates a typical packet in a real time media stream.

33) A packet as claimed in claim 26, wherein the packet is of the same size and user data protocol port range as the standard packet.

34) A packet as claimed in clam 33, wherein the packet is sent over the telecommunication network to emulate the standard packet.